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DEVELOPMENT OF A REMOTE SOUND MONITORING SYSTEM

BY

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RESEARCH REPORT

Submitted in partial fulfillment of the requirements
for the degree of Master of Science in Engineering
in the Graduate Studies Program of
Florida Technological University, 1974.

Orlando, Florida

ABSTRACT

This report explains the development of a sound measuring system for continuous monitoring of sound levels. Dedicated telephone lines are used for data transmission so that sound level data can be monitored at installations far from the area of concern. This allows the sound sensing equipment to be left unattended and thereby greatly reduces the manpower costs of operation. That predetermined data conditions can be detected when they occur and corrective action can be taken immediately at the data receiving installation further justifies the method used.

Background information and general need for such a system are discussed. Growing public concern with noise is creating a demand for more and better sound measurement devices and for more control of unnecessary noises.

For description, the system is divided into functional sections. The sections are described and design considerations are given in order of data flow starting with the sound sensing device, or microphone, and ending with the warning system which alerts the operator when a predetermined signal level has been exceeded. Circuits to detect selected conditions of excess sound and to control the alarm to these conditions are included.

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CHAPTER I

ENVIRONMENTAL CONSIDERATIONS

As public concern with pollution grows, the annoyances and problems of noise are brought more and more into the focus of public attention. Long exposure to intense noise levels can permanently impair hearing while lesser levels, although causing no physical damage, can be very irritating.¹

In response to the rising public awareness and concern, governing bodies are more and more taking steps to combat the problem. Among other things, anti-noise ordinances are being passed, and long range plans are being discussed for locating airports to affect a minimal number of people. At the same time, noise producing industries are taking a hard look at the cost and feasibility of noise reduction.

In noise reduction and control a prime prerequisite would be to ascertain the level of noise in question. Because

¹Arnold P. G. Peterson and Ervin E. Gross, Jr., Handbook of Noise Measurement (7th Ed.; Concord, Mass.: General Radio Co., 1972), p.1.

of the complexity of individual subjective reaction to frequency and level of noise, a simplified system of measuring sound pressure level using A-weighting for relatively nondirectional outdoor sources is widely used.² Figure 1 shows an A-weighted frequency response curve.

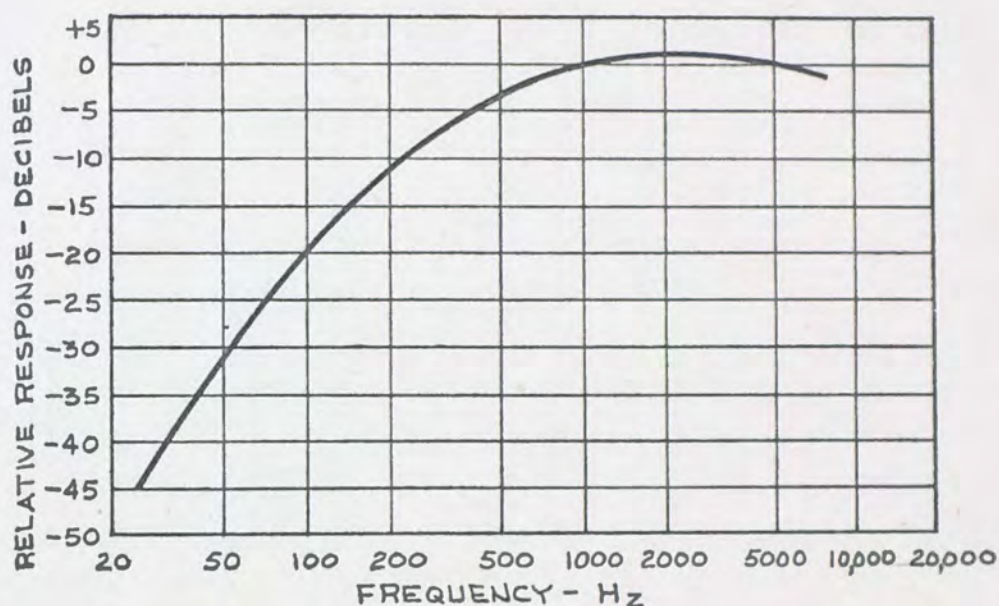


Figure 1.--Frequency response for A-weighting characteristics.

Sound measurements are commonly made with commercially available, portable units which require constant attention for operation and data taking. While inexpensive initially, these units could require large operational costs for long term measurements because of the manpower involved. Some of these instruments can be used with recording equipment such as stripcharts, thereby greatly reducing the need for operator time. However, the data taken in this manner is ex post facto in nature and precludes the monitoring of

²Ibid., pp. 42-44.

the data as it is taken. It should also be noted that as operator surveillance time is reduced, the risk of data loss through unobserved equipment malfunction is increased. Compromises would need to be made in such cases using cost and data importance as factors.

The design of this system is based on the need for constant recording of sound levels at a selected site located a relatively large distance from the sound source. The monitoring equipment could be located at the sound source as shown in Figure 2. The system would require a minimum of maintenance, and would require monitoring only when sound levels are sufficiently high to cause concern. When predetermined sound levels were reached, an annunciator system would alert the system operator who could then take actions deemed necessary depending on the circumstances. The operator would be free for other duties at all other times, which could greatly reduce the manpower costs of system operation. This, of course, would depend upon the nature of the specific function for which the system was used.

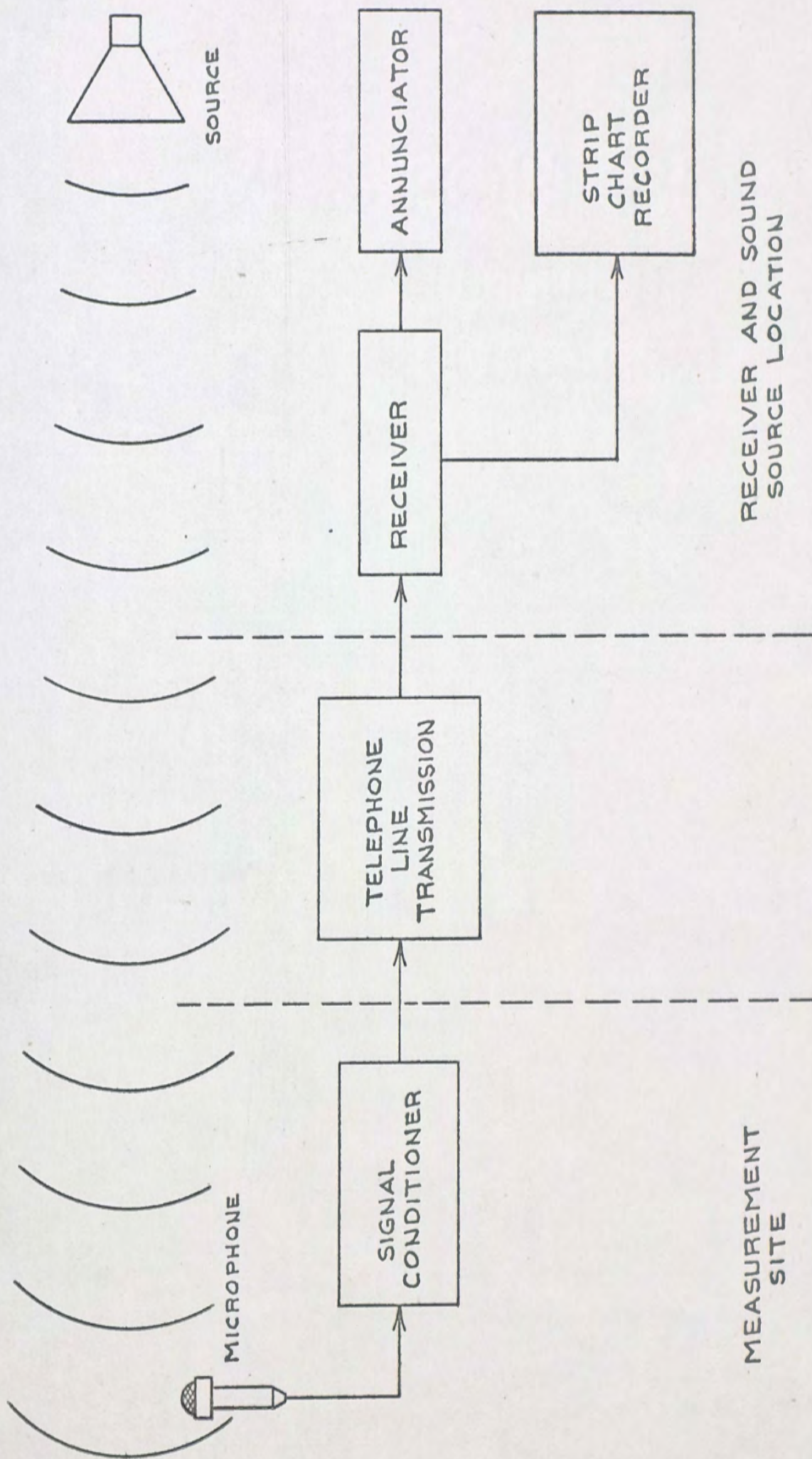


Figure 2.---Block diagram of remote sound monitoring system.

CHAPTER II

DATA TRANSMISSION DESIGN

Choice of Acoustic Instruments

In sound measurements the microphone must certainly be given primary consideration. Choice of type used must be based on accuracy required, frequency response and environmental conditions to which the microphone will be exposed. A B&K Model 4133 microphone with the Model 2619 field effect transistor type preamplifier was selected.

These microphones are of the condenser type, and when used with the FET preamplifier have a dynamic range of approximately 5 to 160 decibels. A calibration certificate, including a complete frequency response curve is furnished with each microphone cartridge. These calibrations are traceable to the National Bureau of Standards, Washington, D. C. The microphones are also well adapted to outdoor use because of their high resistance to humidity and their very small temperature coefficients.³

Sound to DC Voltage Conversion

The signal coming from the microphone preamplifier is a

³B&K Instruments, Inc. Product Data (Booklet 13-003)
(Copenhagen, Denmark) n.d.

low level ac signal analagous in amplitude and frequency to the sound pressure level (SPL) and frequency of the sound detected at the microphone. Because of the low signal level (approximately 4 mV rms at 85 dB SPL) two stages of operational amplifier are used to increase the voltage level before filtering. Variable gain allows setting a predetermined filter output level. In this case, 1 V rms for 85 dB SPL is selected.

Filtering can be tailored to fit the individual needs using any of the available filtering methods. For this application a band pass filter network was accomplished by using commercially available, passive, 6 pole high and low pass filters of the United Transformer Company type. The band width is selected to attenuate all frequencies outside the spectrum of interest in the particular measurements being taken.

As band width is decreased, the ideal situation of measuring only the contribution to the overall level which is made by the specified, far away source is approached. Over long distances, the cutoff frequency of the low pass filter can be determined through the factor of molecular absorption of sound in air.

Attenuation caused by molecular absorption increases with frequency, relative humidity and temperature as well as

distance. For example, Beranek⁴ estimates that over a distance of 10 miles with conditions of 70°F. and 43% relative humidity, sound components with frequency of 600 Hz are attenuated 35 dB. For the same conditions, components at 300 Hz are attenuated only 20 dB. Therefore, if limits of temperature and humidity are known, the highest frequency of concern can be determined for a given distance knowing source level maximums.

A signal analagous to sound level is obtained by rectifying, or detecting, the filter output waveform. This is done with a precision half-period averaging circuit having one active amplifier element. Time constant considerations of the detecting circuit require that the lowest sound frequency passed by the filter be 4 to 5 times higher than the frequency of sound level change to which good system response is needed.

Sound Pressure Level in Decibels

Since sound is normally referred to in decibels, the next step is to convert the low-frequency sound level signal to a logarithmic scale. By doing this, the monitoring equipment is simplified for the operator since the dB readout scale can be linear instead of logarithmic.

Where the data is recorded, as in this case, the recording

⁴Leo L. Beranek, ed., Noise Reduction (New York: McGraw-Hill, 1960), pp. 188-191.

paper can be of linear nature and more easily read.

In certain semiconductor junctions, there is a logarithmic relationship between voltage and current. Over the years, refinements have been made by searching out the junctions that best follow the ideal formula

$$e = E_0 \log_{10} \frac{i}{i_0}$$

and compensating for temperature, range extension, etc. E_0 and i_0 are reference values. Three basic logarithmic configurations that have evolved are shown in Figure 3.

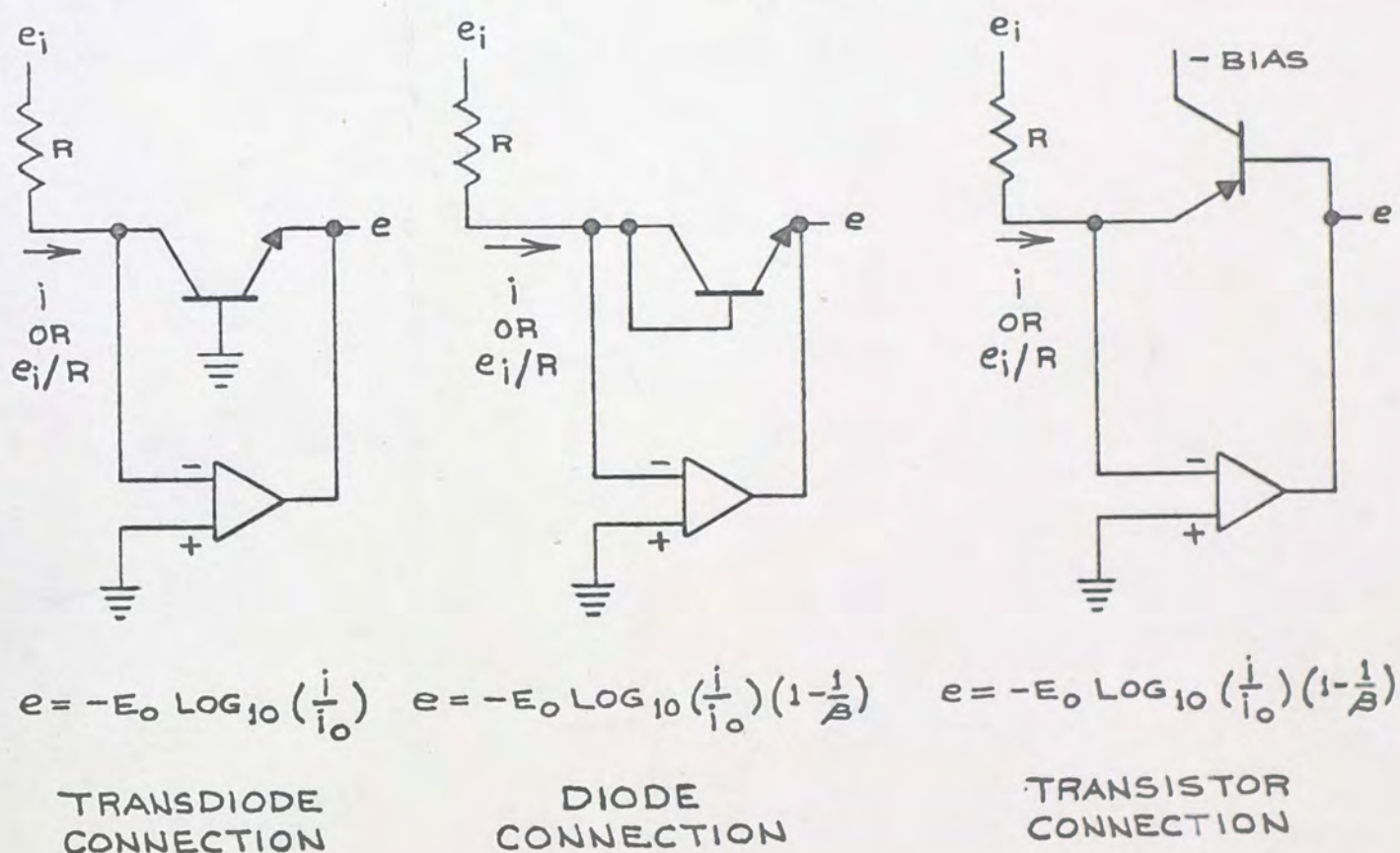


Figure 3.--Basic logarithmic configurations.

The transdiode configuration has the best current resolution, but must be used in single polarity applications.⁵

⁵Alan R. Risley, "Designers Guide to Logarithmic Amplifiers," EDN Magazine, August, 1973, pp. 42-51.

The Analog Devices model AD751N logarithmic module was selected to convert to a logarithmic scale. This module contains two NPN type transistors along with a temperature compensating voltage divider. One transistor element is used in the transdiode configuration while the other transistor element, the compensating voltage divider and an additional operational amplifier are used to establish a constant current corresponding to i_0 in the equation. The voltage relationship is logarithmic as shown in Figure 4.

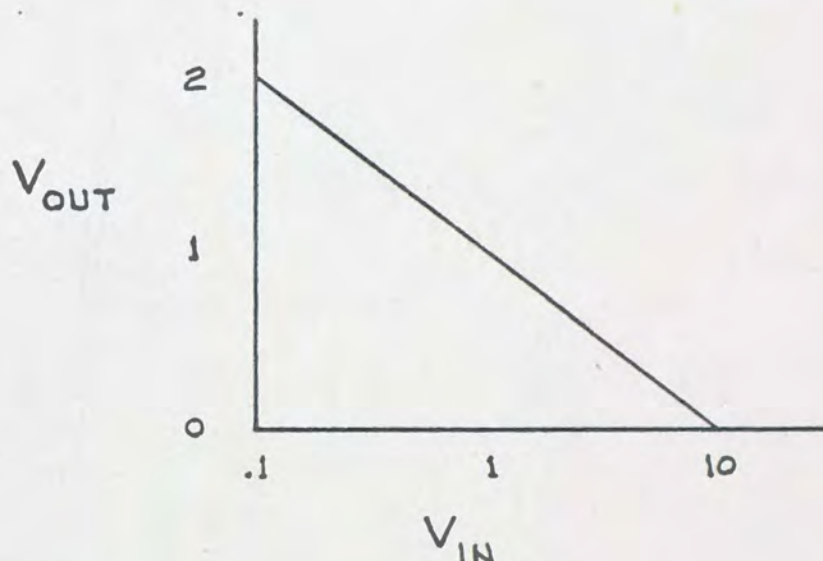


Figure 4.--Logarithmic module input vs. output.

Telephone Line Transmission

One of the needs of the system is the capability of monitoring sound data at sites that are far removed from the monitoring system. The distances between the microphone and the monitoring equipment can be as large as 20-25 miles or even more. Obviously, installation of wires over these large distances is impractical or even impossible. For instance, right-of-way would need to be obtained outside

the limits of the operation. In any case, the cost would probably be prohibitive and line loss, attenuation, etc. would tend to limit the validity of the data unless some scheme were used which did not rely on amplitude accuracy of the transmitted signal.

Radio transmission could be used. Frequency modulation, a common telemetry tool, would allow accurate data reproduction at the receiver, and large distances could be covered depending on the transmitter power capabilities. However, the cost of such equipment could be a problem. A frequency band of operation would have to be assigned and an operating license issued by the Federal Communications Commission.

A third choice, which was chosen for this case, would be transmission by telephone line. Here, for a nominal monthly cost, a circuit can be rented for exclusive use. A pulse transmission circuit with a limit of 30 pulses per second was chosen because of the lower rental rate. Circuits with much higher frequency capability are available at higher rates, but the time response of this system can be as large as several seconds without loss of data usefulness. Therefore 30 pulses per second transmission is adequate.

The requirement at this point was to convert the signal, proportional to sound level in decibels, to pulses which were compatible with the telephone circuit.

For converting the signal to pulses, a voltage to frequency converter was designed. The basic concept is that of a relaxation oscillator: a ramp generator, a comparator and a reset stage. The ramp generator and comparator were devised using inexpensive operational amplifiers and the reset was accomplished by using a PNP transistor. The output frequency band selected was 5-25 Hz. This keeps a respectable 5 Hz above zero and the same below 30 Hz which is the telephone circuit limit. The voltage to frequency converter was set to 10-50 Hz to facilitate the wave shaping that follows. The output, a negative going pulse of very short duration, was then shaped by using a type MC664 master-slave R-S flip-flop. By feeding the pulses into the clock input and feeding back the Q output to both reset inputs (R_1 and R_2) and the \bar{Q} output to both set inputs (S_1 and S_2), the Q output gives a square wave of one half the input frequency. The frequency band of 5-25 Hz is thus obtained.

The telephone circuit requires only an alternately opening and closing of the circuit at the customer input. The circuit is current limited at approximately 62 mA. Isolation from ground is also required. Because of the isolation and relatively high current requirements, a Clare model HGS 1006 mercury-wetted contact relay was used. The relay is capable of up to 100 operations per second, is free from contact bounce and has a rated life

expectancy of more than one billion operations.⁶ These qualities assure a clean waveform and over 1.2 years of life at the maximum frequency of 25 Hz. A 2N3904 transistor, in switching mode, drives the relay coil from the flip-flop output squarewave.

⁶Mercury-Wetted Contact Relays: Manual 201C (Chicago: C. P. Clare & Co., 1964), pp. 4-5.

CHAPTER III

DATA RECEIVER DESIGN

Pulse Signal Conversion

As stated in Chapter II, the telephone circuit requires isolation from ground and other circuit components. To achieve this, another mercury-wetted contact relay is driven by the telephone circuit on the receiving end. A low current Clare model HGSM 1001 relay was selected for this purpose and gives ample isolation for the telephone circuit. In addition to long life and absence of contact bounce, this relay is magnetically biased. This biasing causes the relay to be polarized and prevents response to any negative excursions that might be in the incoming waveform.

The incoming square wave is duplicated on the normally open contacts of the relay and injected into the Schmidt-trigger input of a SN74121 monostable multivibrator. By connecting 15,000 ohms resistance and 2.0 microfarads capacitance to the proper timing pins, an output of constant 20.8 milliseconds duration is obtained. The pulse width is defined by the relationship

$$t_{p(out)} = C_t R_t \log_e 2.^7$$

⁷The Engineering Staff of Texas Instruments Incorporated Components Group. The Integrated Circuits Catalog for

Because the maximum data pulse repetition rate is 25 pulses per second, or a period of 40 milliseconds, the constant pulse width of 20 milliseconds from the multivibrator represents a maximum duty cycle of 50%. Or, if the rate should overscale to the telephone circuit maximum of 30 pulses per second, the duty cycle still would not exceed 60%.

Since amplitude and pulse width are constant (constant energy per pulse) and since the repetition rate is the same as the incoming signal, time integration of the pulses will give a dc voltage level that varies in direct proportion to incoming frequency.

Integration with respect to time is accomplished by using an operational amplifier in an analog integration configuration. Integration occurs over relatively long periods of time (20 millisecond pulse width and minimum of 40 millisecond period), and the liberal response-time requirements of this system allow a time constant $T=ARC$ of 800 milliseconds, where A is the amplifier low frequency gain. Therefore, the integrator operates over the linear portion of the response curve and gives computation accuracies well within the needs of this system.⁸ An adjustable bias voltage to the noninverting input of the

Design Engineers (Dallas: Texas Instruments, Inc. [1973]), p. 6-73.

⁸Jerald G. Graeme, Gene E. Tobey, and Lawrence P. Huelsman, ed., Operational Amplifiers: Design and Applications (New York: McGraw-Hill, 1972) pp. 213-217.

amplifier compensates for output bias errors and is also a source of adjustment for the output signal "zero."

The integrated signal, which is a dc voltage varying with pressure level in decibels, is fed to monitoring devices. These devices consist of a stripchart recorder for continuous data history and a meter relay. The meter relay displays the dB level for monitoring, but more importantly it also acts as a switch which closes when the dB level exceeds a predetermined setpoint level. This switch obviously could be used to control visual and/or audible alarms to alert the operator when this level is exceeded.

The requirements of this system, however, are more complex than merely knowing each time a given sound level is exceeded. In fact, if the sound level were fluctuating at a fairly rapid rate, the alarm system could become a problem. The determination of the type of alarm system to be used is described in detail below.

Annunciator Requirements

The primary objective of a monitoring system of this type is to determine the portion of sound level in the monitored area which is caused by the source in question. Although the sensor filters have been designed to pass only those frequencies which can propagate from the source to the sensor, this does not preclude sound levels in the same frequencies which are generated in the sensor area. Obviously, sound level control at the source will have no

effect on these levels.

This problem becomes less ominous when it is realized that a dB level expresses a ratio between a given magnitude and a reference. Therefore, when noise levels stated in decibels are combined the number of decibels are not added directly but are converted to ratios, added and then converted back to decibels. For example, consider the combining of two ratios X and Y which have been stated in decibels where $X \geq Y$.

$$dB(x) = 20 \log_{10} X \quad \text{and} \quad dB(y) = 20 \log_{10} Y.$$

Rearranging and taking antilogarithms gives

$$X = \log_{10}^{-1} \frac{dB(x)}{20} \quad \text{and} \quad Y = \log_{10}^{-1} \frac{dB(y)}{20}.$$

The values for the ratios X and Y can then be found from a table of logarithms. To find the value in decibels that results from adding X and Y we note that

$$X + Y = X(1 + \frac{Y}{X})$$

and the combined level dB_T is found as

$$\begin{aligned} dB_T &= 20 \log_{10} X(1 + \frac{Y}{X}) \\ &= 20 \log_{10} X + 20 \log_{10} (1 + \frac{Y}{X}). \end{aligned}$$

Therefore, combining Y with X increases the decibel level of X by the amount $20 \log_{10} (1 + \frac{Y}{X})$. Since $X \geq Y$ the maximum increase occurs at $X = Y$ and is $20 \log_{10} 2 = 6$ dB. At higher sound levels this becomes less significant, e.g., a combining of two levels of 80 dB each would result in only 86 dB.

If a history of average or constant sound levels generated in the sensor area can be obtained, allowances can be made and the annunciator setpoint established accordingly.

However, there could be noise levels in the area that are transient in nature as from passing vehicles, construction activities, etc. For these conditions of relatively long duration which can be more or less anticipated from the area history, a delay circuit which allows the setpoint to be exceeded for 120 seconds before alarm conditions are triggered seems adequate.

This, however, could allow repeated source generated sound levels which exceed the setpoint with the only limiting condition being that each period of sound generation not exceed the delay time. To avoid this, alarm conditions arise if the setpoint is exceeded for twenty seconds during each of N thirty-second periods in any ten minute memory time, where "N" is selectable for values of one through ten. A memory unit stores twenty bits of data, with each bit representing a thirty-second period, for a total of ten minutes memory.

Annunciator Circuit Design

As previously stated, the annunciator circuit was designed to alert the operator if either of two conditions exist:

1. The measured sound level exceeds the setpoint continuously for 120 seconds.
2. The level exceeds the setpoint for 20

seconds during each of N thirty-second periods in a ten-minute time frame, where "N" is selectable from 1 to 10.

The heart of the annunciator circuit is a clock having a 0.1 second period. These clock pulses were obtained by using an active relaxation oscillator synchronized to one sixth power line frequency and are negative going pulses. The meter relay is connected in the normally closed configuration so that the output is low when the setpoint is exceeded. These two signals, the clock pulse and the meter relay output, are then used to determine the occurrence of the prescribed conditions and to trigger the warning system when the conditions are met.

To detect the existance of high sound levels that have continued for 120 seconds, TTL counters were used. By properly selecting use of the divide-by-two, divide-by-six and divide-by-ten sections of these counters a divide-by-2400 network was achieved. The negative going clock pulses are conditioned for compatibility with the counter logic by applying the pulse train to both inputs of a 2-input positive NAND gate. The NAND gate then supplies to the counter bank a pulse train of clock frequency which conforms to the requirements of the positive logic counters. The counter output is then a logical 0 for 120 seconds at which time it switches to logical 1. By use of an OR gate (see Figure 5) the counter circuit

is brought to reset condition whenever the meter relay is below setpoint or when the counter output goes to logical 1.

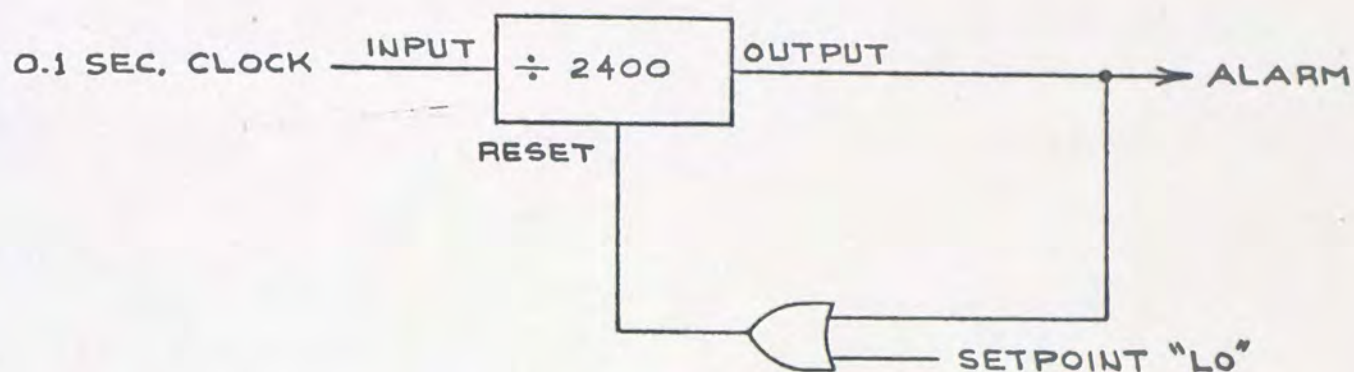


Figure 5.--120 second delay diagram.

This self resetting action creates a pulse, and this pulse, through appropriate conditioning amplifiers, de-energizes a self-holding relay circuit and produces a warning signal until the relay circuit is reset manually. Since the counter is in reset condition any time the meter relay is below setpoint, alarm conditions occur only if the setpoint has been exceeded continuously for 120 seconds.

A second clock with 30 seconds period was then devised by using a divide-by-300 counter to convert the 0.1 second clock pulse to a square wave of 30 seconds period. A negative-edge-triggered monostable multivibrator, with pulse width of 0.1 millisecond, converts this wave to a pulse train. The function of the 30 second clock is given in the following description.

For the second condition of warning, the 0.1 second clock pulse and the meter-relay output are fed into a 2-input

positive NAND gate. This produces an output only when the meter-relay is low, i.e., when the setpoint is exceeded. These pulses are then fed to a divide-by-400 counter. Since this counter is reset only by manual operation or by the 30 second clock which is described above, it accumulates counts and the output changes state after 20 seconds. Reset does occur every thirty seconds, however, and therefore the counter output goes to logical 1 only if 20 seconds of high level sound have accumulated during any 30 second period.

The counter output is fed into a monostable multivibrator with output pulse width of 0.1 milliseconds. This pulse, in turn, is fed into the clock input of a D-type edge-triggered flip-flop. By keeping a constant logical 1 at the D and PRESET inputs, the flip-flop becomes a one-bit memory. A 0.1 millisecond pulse from the monostable multivibrator causes the Q output to switch to logical 1 and remain until the flip-flop is cleared by the 30 second clock. Therefore, if 20 seconds of high signal level accumulates during a 30 second clock period, the flip-flop output goes high and remains until the next 30 second clock pulse. If 20 seconds of high level sound have not accumulated, the flip-flop remains low throughout the 30 seconds of clock period. Figure 6 is a logic diagram, and Figure 7 is a timing diagram showing various conditions controlling flip-flop output.

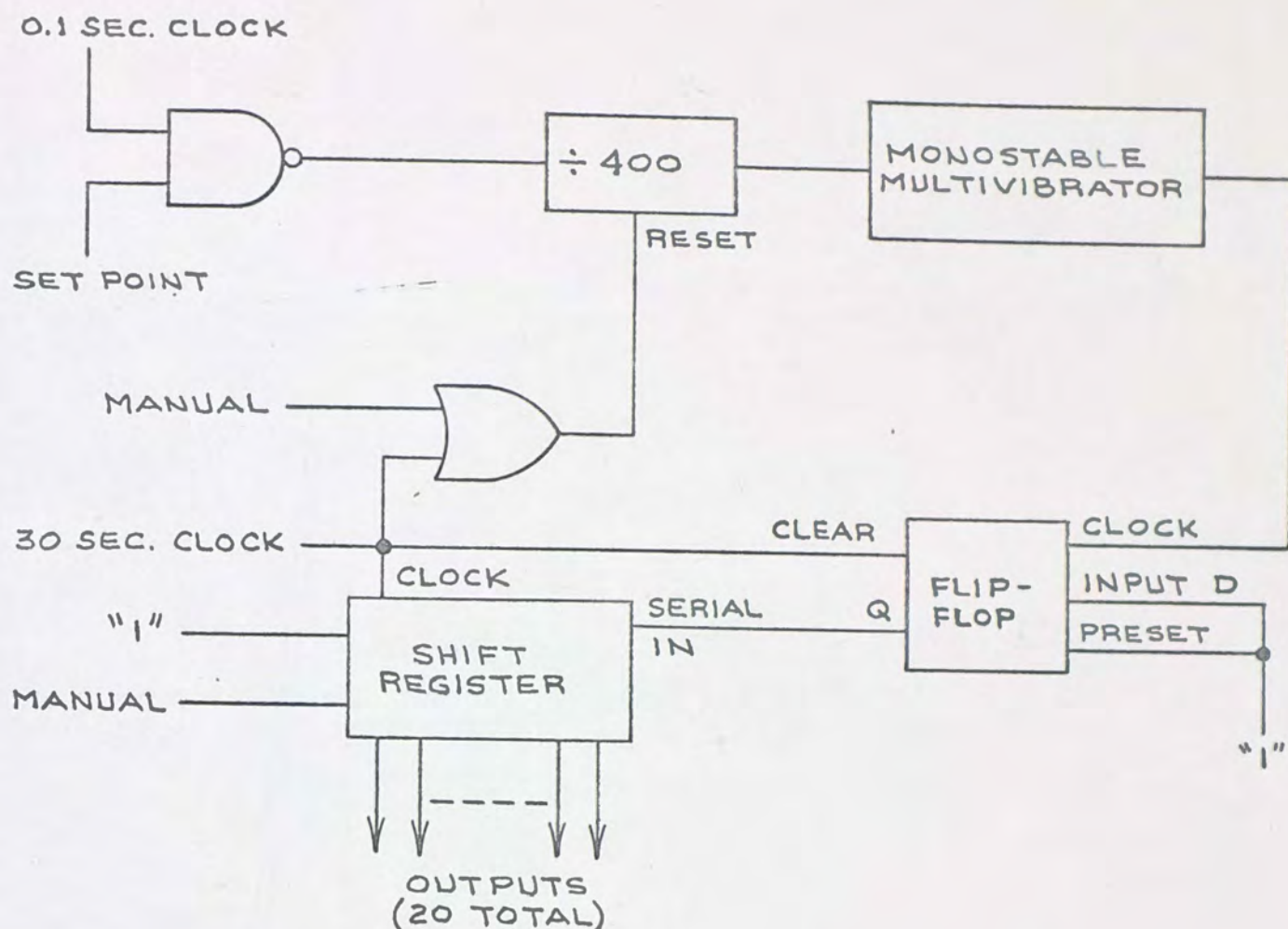


Figure 6.--N-system logic diagram.

For ten minute storage, four 5-bit shift registers were used. By connecting these registers in serial configuration, a 20-bit register was developed. Clocking all registers with the 30 second clock pulse provides storage of bits representing ten minutes of time that is updated each 30 seconds.

Each logical 1 bit in the register represents a 30 second time period in which the setpoint has been exceeded for at least 20 seconds. To determine the number of these bits in storage, a simple summing circuit was selected.

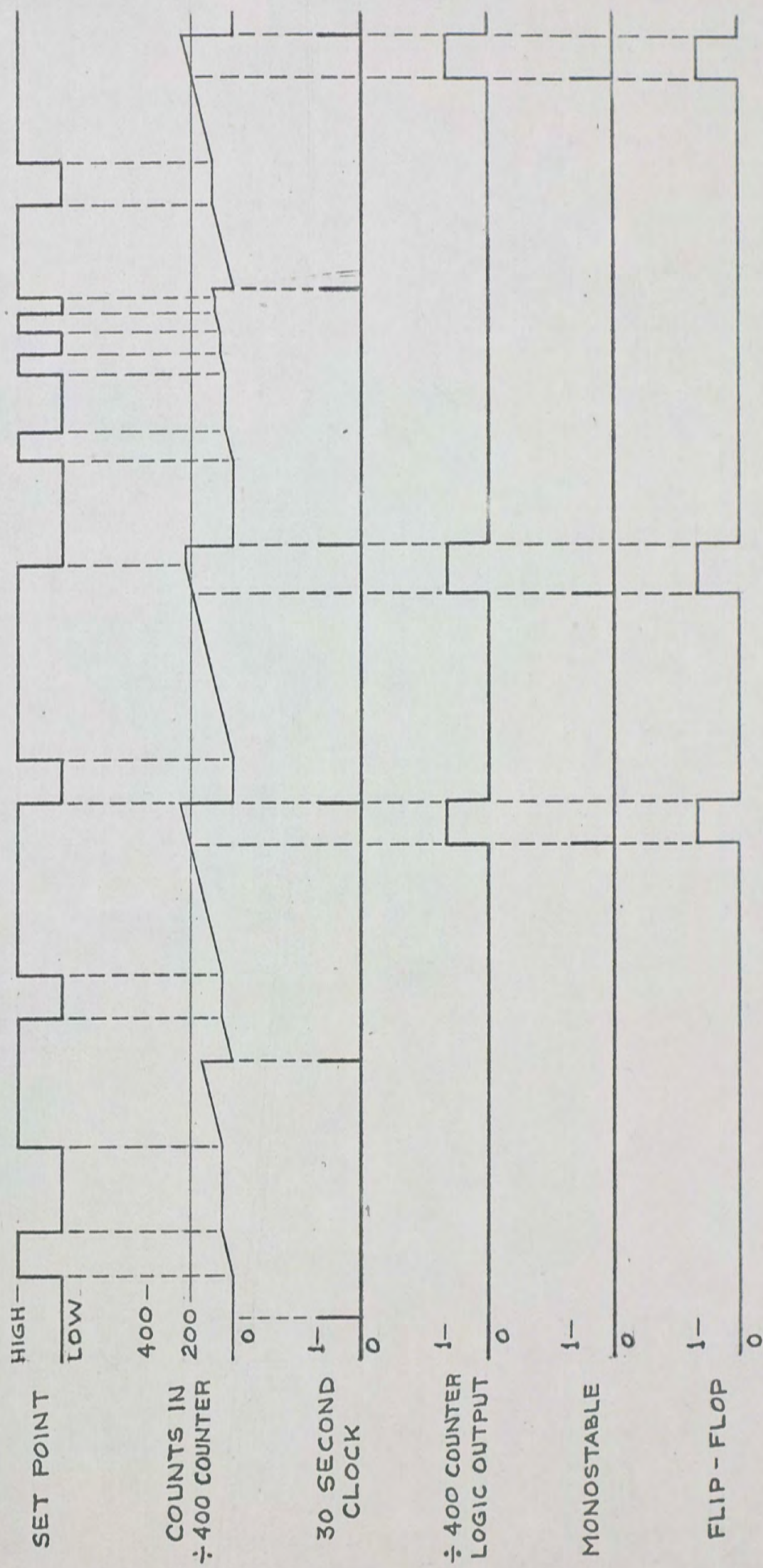


Figure 7.--Timing diagram for N-system.

Each of the twenty register outputs was connected through a 10,000 ohm resistor to the summing junction of a summing amplifier. A 4A741C operational amplifier was used for this purpose. The summing amplifier output then has twenty-one discrete levels representing the possible number of logical 1's from zero through twenty. A rotary switch selects discrete voltage levels from a voltage divider arrangement of resistors. This switch selects "N" which is the number of 30 second periods, each containing 20 seconds of high level, that are required to operate the alarm. Figure 8 shows the output of the selector switch and the output of the summing amplifier going to the non-inverting and the inverting inputs respectively, of an operational amplifier.

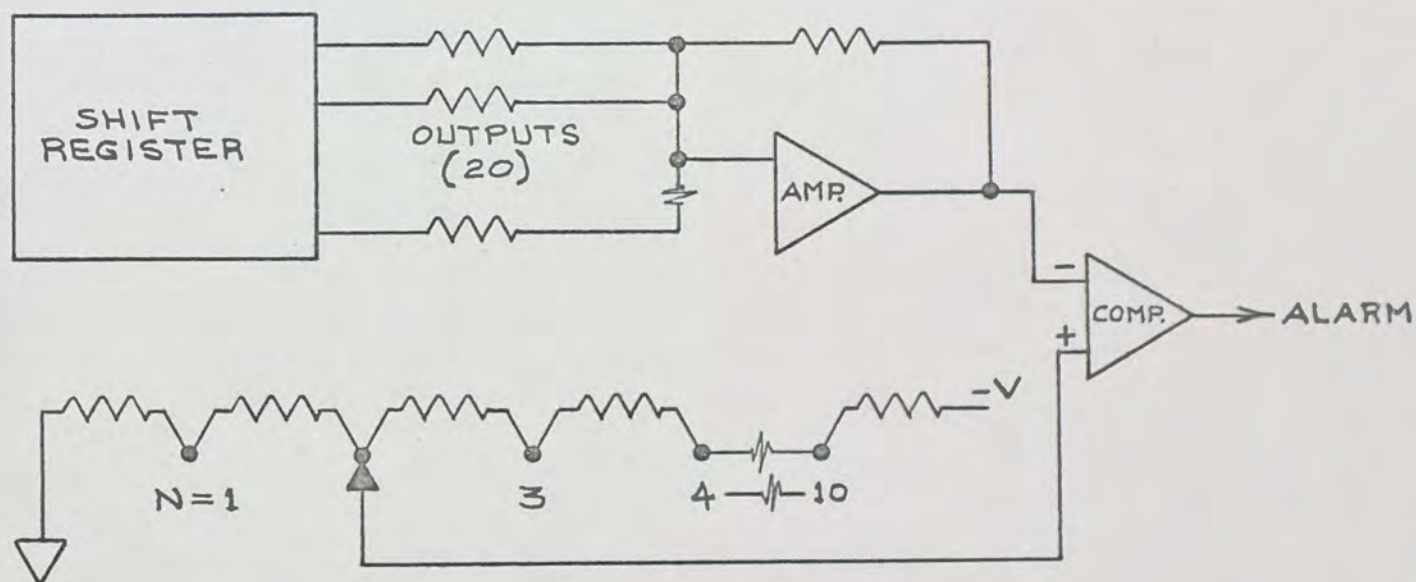


Figure 8.--N-system digital to analog conversion diagram.

The amplifier has no external feedback circuitry and therefore acts as a comparator because of its high gain. When

the number of logical 1's in the register exceeds the "N" setting of the selector switch, the comparator changes state and the alarm sounds. The register can be cleared of all bits, and the system reset, by a manually operated switch.

CHAPTER IV

RESULTS AND CONCLUSIONS

The telephone circuit data transmission technique used in this system allows a virtually unlimited distance between sound sensor and data receiver. Limits would, of course, be imposed by factors such as telephone line leasing costs which would increase with distance and tariff boundary crossing. Sensor location is also limited by availability of telephone and power line service. Cost of installation of these line services also requires that equipment installations be more or less permanent or non-portable.

Several of these systems are in use at jet engine testing installations to help control sound levels in surrounding communities. Annunciator setpoints are determined by ambient sound level history in the sensor area and by engineering studies of sound propagation as well as the human factor.

When the annunciator alarm sounds, the strip chart recording can be viewed to determine the sound level profile, and engine testing levels can be modified to determine their contribution to the immediate overall noise level in the

sensor area. Here, distance to sensor and speed of sound must be taken into consideration. Once the contribution is determined, appropriate actions can be taken. The stripchart data is also valuable for its use in sound propagation studies. Used with time correlated weather data, for example, meteorological effects on propagation can be studied for use in further sound level control efforts.

Although this system was designed for sound level monitoring, it could easily be adapted to other environmental measurements where need for constant monitoring and/or continuous history is justified. Appropriate sensors would need to be selected, and some modification to the transmitter electronics could be required. If considerably greater time response were required, telephone line frequency limits would need to be increased, and high speed switching devices would be needed to replace the mercury-wetted contact relays.

BIBLIOGRAPHY

Peterson, Arnold P. G., and Gross, Ervin E., Jr. Handbook of Noise Measurement. (7th Ed.) Concord, Mass.: General Radio Co., 1972

B&K Instruments, Inc. Product Data. (Booklet 13-003). Copenhagen, Denmark, n.d.

Beranek, Leo L., ed. Noise Reduction. New York: McGraw-Hill, 1960.

Risley, Alan R. "Designers Guide to Logarithmic Amplifiers," EDN Magazine, August, 1973. pp. 42-51.

Mercury-Wetted Contact Relays: Manual 201C. Chicago: C. P. Clare & Co., 1964.

The Engineering Staff of Texas Instruments Incorporated Components Group. The Integrated Circuits Catalog for Design Engineers. Dallas: Texas Instruments, Inc. [1972].

Graeme, Jerald G.; Tobey, Gene E., and Huelsman, Lawrence P., Ed. Operational Amplifiers: Design and Applications. New York: McGraw-Hill, 1972.